AMENDMENTS TO THE CLAIMS

Please cancel claims 7 and 12 without prejudice and accept amended claims 1, 8-10 and 20 as follows:

1. (Currently Amended) A method for separating at least two audio channels recorded using an array of at least two microphones comprising the steps of:

equalizing variances of a first channel and a second channel on a current data frame;

recursively expressing means and variances of mixtures; [[and]]

normalizing the second channel to a variance level substantially similar to a variance of the first channel; and

determining delay parameters by minimizing a cross-covariance between two outputs.

2. (Original) The method of claim 1, wherein on a current block of m data samples $x_j(t)$, $1 \leq t \leq m$ $1 \leq j \leq 2$, and index k, a current block mean $\overline{x}_j$ can be determined according to:

$$\overline{x}_j = \frac{1}{m}\sum_{t=1}^{m} x_j(t)$$

3. (Original) The method of claim 1, wherein a running mean $\overline{x}_j^{(k-1)}$ can be updated by:

$$\overline{x}_j^{(k)} = (1-\beta)\overline{x}_j^{(k-1)} + \beta\overline{x}_j$$

where β is a learning rate.

4. (Original) The method of claim 1, wherein a current block variance Var_j is determined according to:

$$Var_j = \frac{1}{m}\sum_{t=1}^{m} | x_j(t) - \overline{x}_j^{(k)} |^2$$

5. (Original) The method of claim 1, wherein a running variance $v_j^{(k-1)}$ is updated by:

$$v_j^{(k)} = (1-\beta)v_j^{(k-1)} + \beta Var_j$$

6. (Original) The method of claim 1, wherein the step of normalizing the second channel further comprises normalizing an average energy to be similar to an average energy of the first channel according to:

$$\hat{x}_2 = \sqrt{\frac{v_1^{(k)}}{v_2^{(k)}}} x_2$$

7. (Cancelled)

8. (Currently Amended) The method of claim [[7]] <u>1</u>, wherein the cross-covariance between the outputs is expanded as:

$$R_{y_1y_2}(\tau) = R_{x_1x_1}(d_1 - d_2 + \tau) - R_{x_1x_2}(d_2 - \tau) - R_{x_1x_2}(d_1 + \tau) + R_{x_2x_2}(\tau)$$

where $R_{x_ix_j}$ is the cross-correlation between x_i and x_j, $1 \le i,j \le 2$.

9. (Currently Amended) The method of claim [[7]] 1, further comprising the step of determining sub-unit-delayed versions of cross-correlations, wherein the delay parameters are determined for a number of lags L.

10. (Currently Amended) A system for separating two audio channels recorded by an array of microphones comprising:

a calibration module for normalizing gain levels between a plurality of channels on each of a plurality of date frames, wherein each data frame is expressed in terms of time, wherein the calibration module compensates the plurality of channels for attenuations at the microphones in a time domain; and

a delay parameter estimation module for accepting an output of the calibration module comprising the normalized and compensated channels, and estimating a delay parameter for a plurality of data frame sizes over a plurality of lag times, and sorting delays to generate corresponding source separated outputs.

11. (Original) The system of claim 10, wherein the source separated outputs of the delay parameter estimation module are output in real-time.

12. (Cancelled)

13. (Original) The system of claim 10, wherein the delay parameter determines relative delays of arrival of wave fronts at each microphone.

14. (Original) A method for separating at least two audio channels recorded using an array of at least two microphones comprising the steps of:

constraining a mixing model of the at least two audio channels in a time domain to direct path signal components;

defining a plurality of delays with respect to a midpoint between microphones, wherein delays depend on the distance between sensors and the speed of sound;

inverting a mixing matrix, corresponding to the mixing model, in the frequency domain; and

compensating for a plurality of true fractional delays and attenuations in the time domain, wherein values of the delays and attenuations are determined from an output decorrelation constraint.

15. (Original) The method of claim 14, further comprising the step of estimating a complex filter for each microphone, wherein the complex filters define the mixing model.

16. (Original) The method of claim 14, wherein the mixing matrix corresponding to the mixing model comprises two delay parameters and two parameters corresponding to the speed of sound.

17. (Original) The method of claim 14, wherein the output decorrelation constraint is a function of two unknown delays and unknown scalar coefficients.

18. (Original) The method of claim 17, wherein the unknown scalar coefficients are attenuation coefficients substantially equal to one.

19. (Original) The method of claim 14, further comprising the step of imposing a minimum variance criterion for a reverberant case over all linear filtering combinations of X_1 and X_2.

20. (Currently Amended) A program storage device readable by machine, tangibly embodying a program of instructions executable by the machine to perform method steps for separating at least two audio channels recorded using an array of at least two microphones, the method steps comprising:

equalizing variances of a first channel and a second channel on a current data frame;

recursively expressing means and variances of mixtures; [[and]]

normalizing the second channel to a variance level substantially similar to a variance of the first channel; and

determining delay parameters by minimizing a cross-covariance between two outputs.